QUALITY OF SERVICE ENABLED ROUTING FOR VIDEO STREAMING IN SOFTWARE DEFINED NETWORK

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2020

QUALITY OF SERVICE ENABLED ROUTING FOR VIDEO STREAMING IN SOFTWARE DEFINED NETWORK

by

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Thesis submitted in fulfilment of the requirements for the degree of Doctor of Philosophy

January 2020

ACKNOWLEDGEMENT

I would like to take this opportunity to convey my sincere thanks and gratitude to my supervisor Yung-Wey Chong, for all help and valuable guidance provided to me throughout my period of research. I consider myself privileged to have had the opportunity to work under her guidance. I'm also grateful to my co-supervisor Asso.Prof Dr. Selvakumar Manickam to help and support me

Likewise, I am also grateful to Professor Dr. Rosni Abdullah as the Director of National Advanced IPv6 Center (NAv6) for providing workspace and equipments for me to perform the experiments as well as to write this thesis.

I also acknowledge and appreciate the love and support to my parents, wife, brother, sister and son. I thank them for their support, understanding and encouragement during every step of my study and writing thesis.

A special thank to my friends and colleagues that has provided me with moral support that enabled me to be successful in my life.

Balasubramanian Nathan

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LIST OF ABBREVIATIONS

AF	Assured Forwarding
AMRCC	Adaptive Multipath Routing for Congestion Control
API	Application Programming Interface
CS	Class Sector
DABS	Delay Aided Bandwidth Search
DiffServ	Differentiated Service
DPID	Data Path Identifier
EF	Expedited Forwarding
FEC	Forwarding Equivalent Class
HDVC	High Definition Video Coding
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
InServ	Integrated Service
IP	Internet Protocol
JSON	Java Script Object Notation
LARAC	Lagrangian Relaxation-based Aggregated Cost Routing
LDS	Link Discovery Service
LLDP	Link Layer Discovery Protocol
LSP	Label Switched Paths

LSR	Label Switch Routers
MADSWP	Maximally Disjoint Widest Paths
MDC	Multiple Description Coding
MPLS	Mulpi Protocol Label Switching
MRVO	Multipath Routing for Video Streaming in OpenFloW Based Network
NI	Northbound Interface
NOS	Network Operating System
OFDP	OpenFlow Discovery Protocol
PHB	Per Hop Behaviour
PSNR	Peak-Signal Noise Ratio
QoS	Quality of Service
QRVO	QoS-enabled Routing for Video Streaming in OpenFlow-based Network
REST	Representational State Transfer
RSVP	Resource Reservation Protocol
RTP	Real-Time Transport Protocol
SDL	Simple DirectMedia Layer
SDN	Software Defined Networking
SI	Southbound Interface
SP	Shortest Path
SVC	Scalable Video Coding
SWP	Shortest Widest Path

TCP	Transmission Control Protocol
UDP	User Datagram Protocol
VOIP	Voice Over Internt Protocol
WSP	Widest Shortest Path

PENGHALAAN BERBANTU KUALITI PERKHIDMATAN UNTUK PENSTRIMAN VIDEO DALAM RANGKAIAN TERTAKRIF PERISIAN

ABSTRAK

Internet telah menjadi titik pertumpuan untuk data, suara, audio dan video. Ianya dengan pesat menjadi platform pilihan untuk tujuan komunikasi dan penghantaran kandungan,, membawa kepada keruntuhan rangkaian telefon PSTN konvensional dan televisyen daratan dan satelit.Penstriman video sedang mengalami pertumbuhan yang amat pesat dan mewakli lebih daripada separuh trafik Internet dan dianggarkan akan mencecah hamper 82% dalam masa 3 tahun akan datang. Penstriman video and kandungan lain yang peka masa menggunakan UDP (berbanding dengan TCP) kerana protocol ini memberikan QoS yang lebih baik dari segi tranmisi yang pantas dan efisien. Pelbagai arkitektur QoS telah dicadangkan untuk meningkatkan lagi kualiti transmisi video seperti IntServ dan MPLS. Baru-baru ini, Rangkaian Bertakrifkan Perisian (SDN), suatu arkitektur pemayaan rangkaian, telah memperkenalkan kemungkinan untuk lebih menambah-baik QoS, terutamanya bagi penstriman video berkod pelbagai penerangan. Dalam rekabentuk semasa, video jenis tersebut ditransmisikan melalui satu laluan yang boleh menyebabkan QoS terjejas. Akibat bentuk video MDC, penerangan video tersebut boleh diasingkan dan dihantar melalui beberapa laluan denagn menggunakan rekabentuk dan kebolehan SDN. Kaedah-kaedah yang wujud sekarang tidak mengambilkira kebingkasan ralat dalam menentukan laluan pelbagai. Dalam transmisi MDC, pautan tak bercantum secara maksimum digunakan untuk menghantar paket menyebabkan ianya kurang efisien. Oleh itu, tesis ini mempersembahkan penghalaan berkebolehan QoS untuk penstriman video dalam rangkaian berasaskan OpenFlow (QRVO). QRVO mempersembahkan rekabentuk dua-lapisan iaitu pengurus

sumber untuk secara terus-menerus memantau rangkai dan mendapatkan data berkaitan rangkaian dan pemilih laluan untuk mengira dan mengenalpasti laluan terbaik untuk menghantar video tersebut. Proses ini dilalukan secara layang dan keputusan dimasukkan ke dalam pengawal OpenFlo yang seterusnya akan mengubah rangkaian tersebut. QRVO telah menambak-baik truput sebanyak 0.95%, kelewatan hujung-ke-hujung sebanyak 4.56% dan kehilangan paket sebanyak 0.66% dalam senario laluan tak bercantum secara maksimum dan juga memberikan truput yang lebih baik iaitu sebanyak 0.65%, end-to-end delay by 6.33% and packet loss by 0.07% dalam senario laluan optimum. Manakala, dalam senario satu laluan, keputusan menunjukkan prestasi yang hampir sama. Perbandingan hipotetikal dengan MRVO juga menunjukkan QRVO mempunyai kelewatan hujung-ke-hujung yang lebih rendah. Kesemua keputusan menunjukkan dengan jelasnya QRVO yang dicadangkan telah menambah-baik QoS untuk video MDC dalam pelbagai aspek.

QUALITY OF SERVICE ENABLED ROUTING FOR VIDEO STREAMING IN SOFTWARE DEFINED NETWORK

ABSTRACT

The Internet has become the point of convergence for data, voice, audio and video. It is fast becoming the preferred platform for communication and content delivery, eventually leading to the demise of conventional PSTN telephone network and terrestrial/satellite television. Video streaming is experiencing an unprecedented growth it currently accounts for more than half the Internet traffic and is expected to reach 82% in the next 3 years. The transmission of video and other time-sensitive content uses UDP (as opposed to TCP) because this protocol improves quality of service (QoS) in terms of providing fast and efficient transmission. Various types of QoS architecture have been proposed and deployed to further improve the quality of video transmission for example, integrated service, differentiated service and multiprotocol label switching. Recently, software-defined networking (SDN), a network virtualisation architecture, has introduced the possibility of further improving QoS, especially for multidescription coded video streaming. In the current design, such videos are transmitted over a single path that can lead to deteriorated QoS. Owing to the nature of multiple description coding (MDC) videos, the descriptions of the video can be separated and sent over multiple paths by taking advantage of the SDN design and features. Existing methods do not consider the best use of error-resiliency when deciding the multipath. In MDC transmission, a maximally disjoint link is used to send the packets, thus rendering it being less efficient. Hence, this thesis presents a QoS-enabled routing for video streaming in an OpenFlow-based network (QRVO). QRVO presents a two-tiered design with a resource manager to continuously and periodically monitor

the network and collect network vitals and a path selection to calculate and identify the best path to transmit the video. This is done on the fly the decision is pushed to the OpenFlow controller which changes the network accordingly. QRVO provided an improved throughput by 0.95%, an improved end-to-end delay by 4.56% and an improved packet loss by 0.06% in maximally disjoint path scenario, QRVO also provided an improved throughput by 0.65%, an improved end-to-end delay by 6.33% and an improved packet loss by 0.07% in the optimal path however in single-path scenarios, it showed almost similar performance. Hypothetical comparison with a multipath routing algorithm for multidescription video streaming over an OpenFlow-based network (MRVO) showed that QRVO had a lower end-to-end delay. The results clearly indicated that the proposed QRVO approach improved the QoS of MDC videos in various aspects.

CHAPTER 1

INTRODUCTION

1.1 Background

The use of Internet Protocol (IP) based multimedia has increased so enormously that it has now dominated the overall Internet traffic, resulting in attention from industry and academia. Global IP video traffic has increased tremendously that Cisco (CISCO Visual Networking Index) (CISCO, 2017)

projected that the number of IP video users worldwide will reach 1.9 billion by 2021 and it will take an individual more than 5 million years to watch online video. As shown in Figure 1.1,multimedia traffic such as video is expected to reach 82% of all Internet traffic by 2021. Usually these applications require stable network resources with minimal delay and packet loss. However, existing network infrastructure is based on best-effort data transmission that does not guarantee quality of service (QoS).

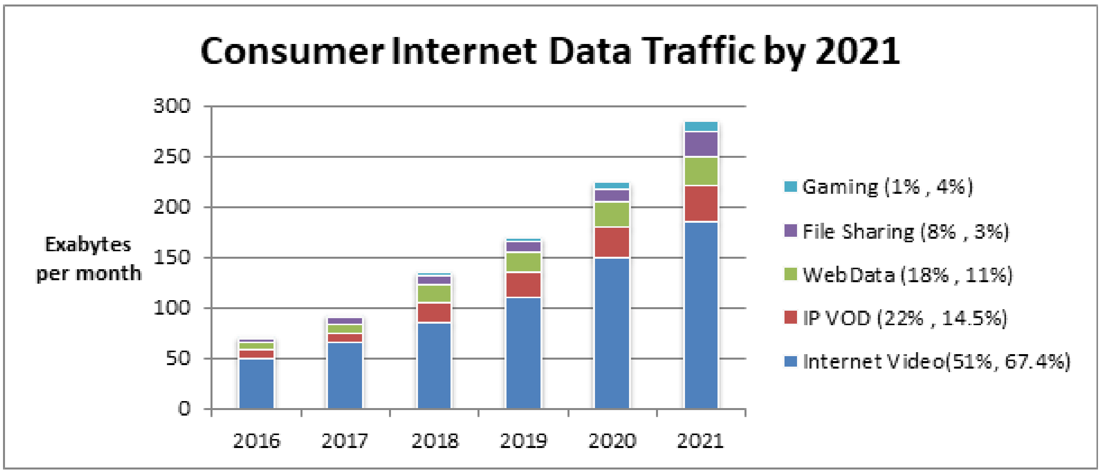

Figure 1.1: Consumer Internet Data Traffic by 2021(Exabytes per Month)(CISCO, 2017))

To support the massive traffic requirement of multimedia, the Internet Engineering Task Force (IETF) proposed two QoS architectures namely integrated services (IntServ) (Braden, Clark, & Shenker, 1994) and differentiated services (DiffServ) (Blake et al., 1998). IntServ utilises resource reservations such as bandwidth to guarantee QoS whereas DiffServ specifies the scheduling mechanism such as priority queuing in managing data traffic. Another method that can provide better routing is multi-Protocol label switching (MPLS), a layer 2.5 technology that uses labels to make packet forwarding decisions. IntServ, DiffServ and MPLS are based on traditional network architecture that operates using closed and proprietary software on switches and routers. One downside of traditional network architecture is the difficulty in designing innovative solutions to cater to the increasing bandwidth-hogging multimedia content.

Software defined networking (SDN) changes the way in which networks are designed and managed. SDN is an emerging architecture that is intended to shape the future of networking to support the enormous growth of data traffic across the Internet. In SDN, network control is directly programmable where the applications and network services are abstracted from the underlying infrastructure. The control plane in SDN has direct control on the network state of the data plane through a prominent application programming interface (API) such as the well-defined OpenFlow (Open Networking Foundation, 2015). Although the future of SDN in providing better QoS has been very promising, there are several research gaps that need to be addressed in order to ensure high quality performance.

1.2 Problem Statement

One of the criteria when designing routing algorithm for a multiple description coding (MDC) enabled video transmission system is selecting path that maximises quality in terms of throughput and end-to-end delay. A method used in the link-state routing protocol is by calculating the probability of packet loss and playback deadline missing using Markov chain process. Probabilities related to the link state are measured by packet drop counts and link speed. Probabilities related to the playback deadline missing are computed by considering queuing delay. The Markov chain process is adopted to measure the queuing delay which estimates the service time of the nodes on which the description may be transmitted.It estimates that two link states such as congested or noncongested could indicate whether packets are always transmitted successfully or characterised by any delay distribution. (Thi, Huynh, & Hwang, 2015).

In this case, to find the optimal paths, the probability calculation of packet-loss and playback deadline that is used to select a link for one description is not sufficient when it utilises a common link for both descriptions in a transmission.

In existing mechanisms, the estimated end-to-end distortion did not consider the available bandwidth parameter that is required to utilise the common links when implementing the partially disjoint routing. (Begen, Altunbasak, Ergun, & Ammar, 2005).

Furthermore, in a path selection process, the existing mechanism emulates the multipath transport that assumes the availability of two statically independent paths between server and client. In this case, the study (Begen et al., 2005) demonstrates that maximally disjoint routing does not necessarily make the best use of the error-resiliency properties of MDC and total link disjoint paths are rarely available between two end hosts. In this case, one of the key mechanisms required for incorporating MDC into routing algorithms is the ability to efficiently select multiple paths between the server and client.

When the mechanism considers only nonshared links for two descriptions, the packets arrive at different times at each node. When any one of the descriptions does not arrive within the playback deadline period, this will affect the quality of the video when the shared link path exists. On the other hand, when the descriptions use shared links, the transmission time is approximately the same for both descriptions until it reaches the disjoint link node.

1.3 Objectives

The overall objective of this thesis is to enhance the QoS of multimedia traffic in terms of bandwidth utilisation, delay, and playback deadline. To achieve the specific objectives, they are defined as follows

1. To propose a QoS-enabled routing mechanism for video streaming in an OpenFlow based network.

2. To propose a modified Dijkstra's algorithm and use it in a source routing-based algorithm so that multimedia traffic can be routed in multiple paths simultaneously.

3. To analyse the effect of the proposed QoS enabled routing mechanism in terms of bandwidth utilisation, delay and playback deadline using emulation.

1.4 Scope

The scope of this work is limited to SDN for real-time traffic, QoS in terms of evaluating the path under network conditions and performance in terms of data rate and delay at the receiver end.

In this thesis, the algorithm is implemented in MDC video with two descriptions. This is because in a multipath routing algorithm, more than two partially disjoint paths are rarely available. The increase of partially disjoint paths may increase the computational complexity (Begen et al., 2005)

In general, using more descriptions and paths will increase the robustness to packet losses and path failures. However, more descriptions may increase the video bit rate for the same video quality (Mao et al., 2006).

The study (Setton, Liang, & Girod, 2003) demonstrates that the most significant performance gain is achieved when the number of descriptions increases from one to two, with only marginal improvements achieved for further increases in number of descriptions.

Further, the scope is limited to the video streaming scenario in which long buffers in streaming make it easier to adapt the available bandwidth, because there is more room for error and adaptations do not need to be as quick as for video conferencing.

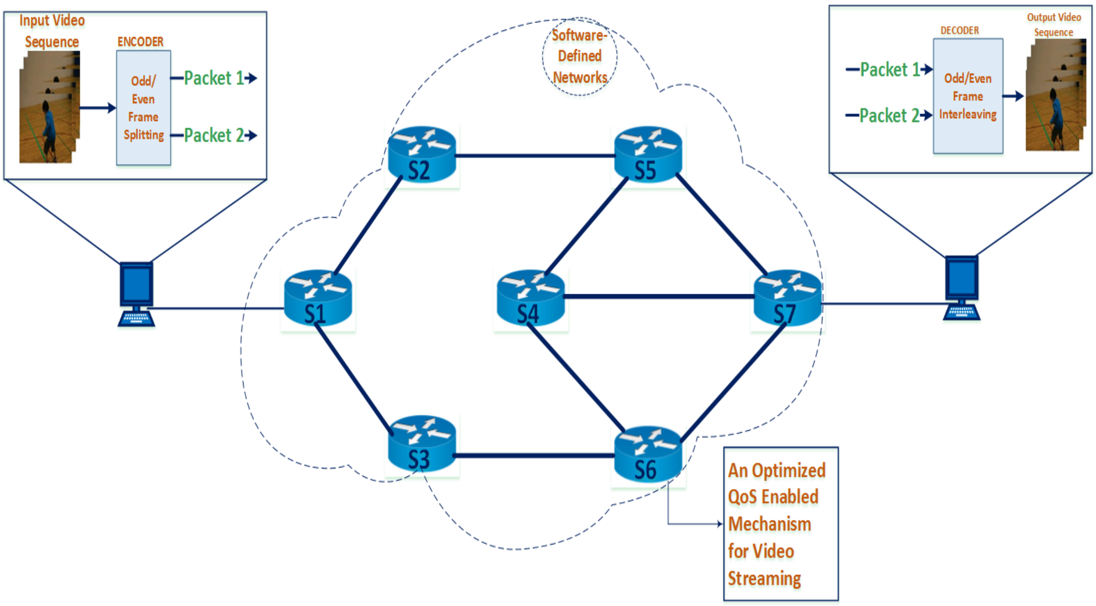

Figure 1.2: Thesis Scope

1.5 Research Methods

To achieve the main objectives as stated in Section 1.4, all the accomplishments of this research can be divided into five important phases. Figure 1.3 illustrates a comprehensive research method of discussion.

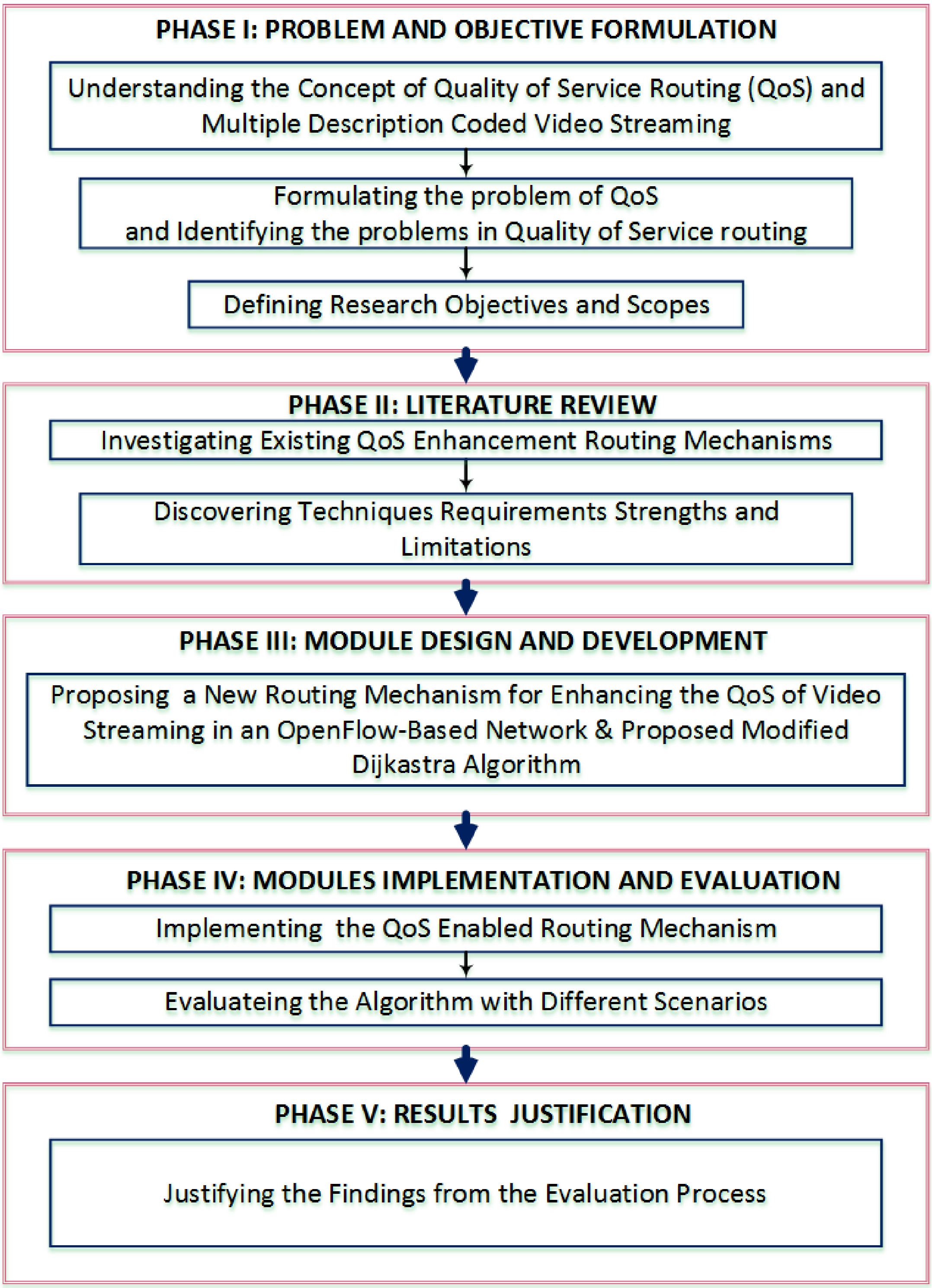

Figure 1.3: Research Methods

1.6 Contributions

There are two main contributions from this thesis:

- A new routing mechanism to enhance QoS video streaming in an OpenFlow-based network. To improve the QoS of video streaming applications, intelligently evaluate the network conditions and utilise the multiple paths during the streaming process. By integrating the multipath prediction, the delay time of the streaming process is reduced which will improve the performance of the video streaming.
- A new formulation of an optimisation problem for routing with MD coded video and the modified Dijkstra's algorithm. Integrate the modify Dijkstra algorithm that enables two routes simultaneously for a seamless streaming process.

1.7 Thesis Organization

This thesis is organised into six chapters.

Chapter 1 presents the preamble of the thesis. It starts by presenting a background discussion of video streaming and QoS routing for real-time applications. In addition, the problem statements, objectives and contribution of the research are defined.

Chapter 2 extensively covers the history and concept of QoS routing and MDC. Current work and related work in QoS routing are explained. The strengths and weaknesses of existing work are also highlighted.

Chapter 3 presents the proposed QoS-enabled routing mechanism for MDC video streaming and its architectural design modules. The new algorithm for QoS routing is explained in detail.

Chapter 4 shows the implementation details of the proposed mechanism. Several scenarios with different datasets are provided. The metrics used to evaluate the performance of the proposed mechanism are discussed.

Chapter 5 presents an in-depth analysis of the results and key findings of the proposed mechanism.

Chapter 6 summarises the work of the thesis. Future work for further research is also outlined in this chapter.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction

This chapter introduces the work related to SDN, QoS requirements of video streaming applications, video streaming technology, routing algorithms and principles for QoS. In addition, it discusses the most related work in terms of QoS-enabled routing mechanism strengths and weaknesses in an OpenFlow-based network. The thesis interest is depicted in Figure 2.1.

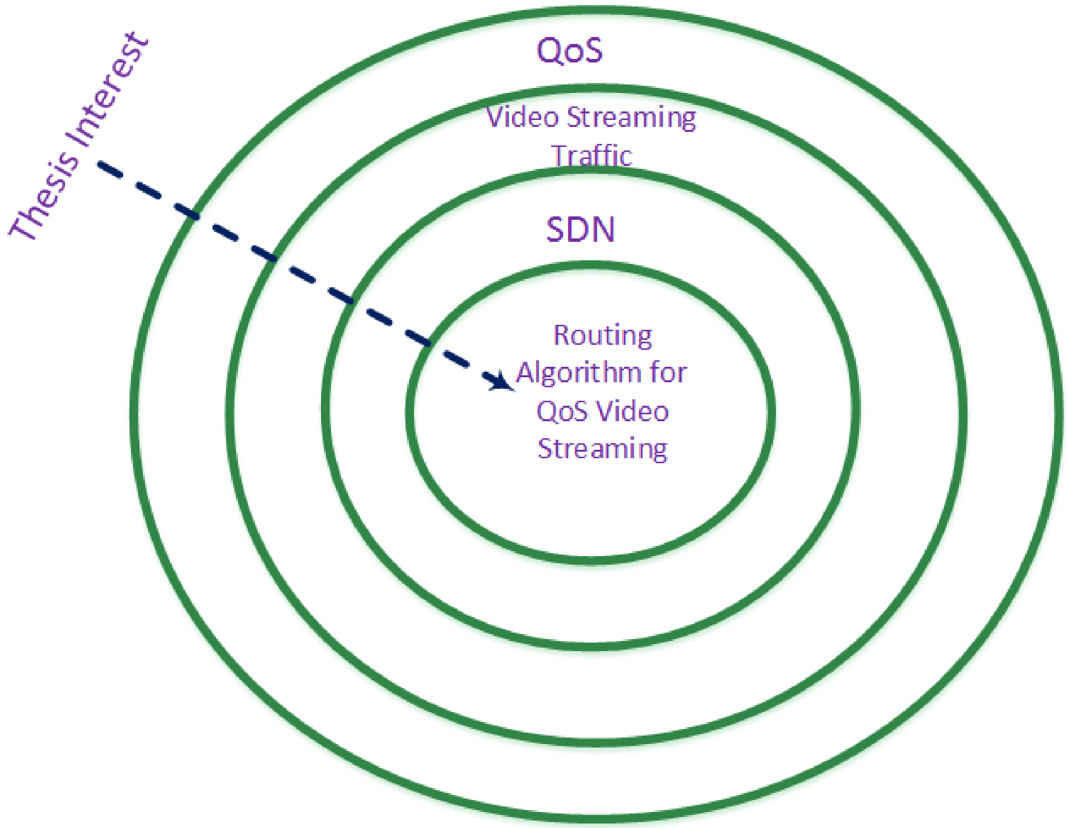

Figure 2.1: Thesis Interest

2.2 Quality of Service

QoS is important for network traffic to achieve optimal overall performance. There are various methods and standardisations to prioritise real-time and interactive traffic, taking into consideration that these do not kill the flow of other types of traffic. The practical use of QoS includes but is not limited to the efficient use and control over resources, tailored services and managing the coexistence of multiple mission-critical applications.

The basic concept of QoS is to manage network traffic to ensure that flows are properly controlled based on their priority by adjusting parameters such as bandwidth, jitter and latency to avoid congestion or mitigation if congestion is unavoidable. QoS is generally processed by identifying and marking the specific flows with levels of priority to achieve the desired throughput. The flows are usually put into queues with the different priorities specified.

Network management systems are becoming more complex and sophisticated in order to support continuously progressing online applications and QoS needs. QoS is an essential mechanism requirement for online and real-time applications, such as video streaming, video-on-demand, online interactive gaming, video conferencing, and virtual collaborative environments, to ensure high-quality performance. The requirement of the QoS connection of the network is mainly dependent on link, path and tree constraints. The link constraint identifies a restriction for the use of the link. The bandwidth constraint of a connection requires the path composed by the link to have free bandwidth available. The path constraint identifies the end-to-end QoS needed on single or multiple paths. The tree constraint identifies to the QoS the need for a whole multicast tree. The unicast and multicast connection requires that the longest end-to-end delay does not exceed the upper bound from the sender to any receiver in the tree constraints (S. Chen & Nahrstedt, 1998).

The selection of the network routes of the QoS routing with adequate resources is determined by requested parameters. In video applications, several parameters such as bandwidth, end-to-end delay, delay variation and reliability are taken into consideration in the QoS optimisation process. To guarantee QoS, routing optimisation, which will satisfy these QoS requirements, is essential. The major components of this implementation include the routing protocol and the routing algorithm. Recent network state information with essential QoS parameters of the network is collected by the routing protocol. A routing algorithm that is capable of finding optimal routes is paramount in order to satisfy QoS requirements. A feasible path that has adequate residual resources such as the unused path is to ensure the QoS requirements of a connection. (W. C. Lee, Hluchyi, & Humblet, 1995). The optimisation of resource utilisation is considered by QoS routing algorithms and it is measured by abstract metric cost. The cost of a link can be measured with buffer function or bandwidth utilisation and the path cost is measured by the sum of all links on the path. The path with the lowest cost is considered to be optimal and is calculated from all feasible paths (Adami, Donatini, Giordano, & Pagano, 2015) (Bari, Chowdhury, Ahmed, & Boutaba, 2013).

There are several challenges and problems that must be considered in QoS routing. Distributed applications such as VoIP and online gaming have disparate QoS requirements on bandwidth, packet arrival delay, delay jitter and packet loss. Because of these various constraints, issues with routing are most often difficult to address. In addition, any future network that handles converged services would potentially be carrying both QoS and best effort traffic, leading to complications in optimisation. The state of any network is dynamic in nature because it is prone to unpredictable changes in load, ingress and egress connection rate and link status (W. C. Lee et al., 1995). Providing high QoS for multimedia applications such as conversational video-over-IP and streaming stored video is always a major concern over the current Internet.

A variety of techniques have been proposed to achieve a preferable QoS including overprovisioning, buffering, traffic shaping, the bucket algorithm and resource reservation. The IETF explored several QoS architectures such as IntServ and DiffServ. Besides, MPLS is a mechanism that allows ultrafast routing.

2.2.1 Integrated Services

IntServ is an architecture that provides the technique to deliver end-to-end QoS for individual flows. To establish and maintain a QoS for a particular flow, it uses the resource reservation and admission control approach by means of the Resource Reservation Protocol. Using this protocol, the IntServ explicitly signals the QoS requirements of user application's traffic with the devices in the end-to-end path over the network. In end-to-end signalling, IntServ requires various functions such as admission control, classification, queuing, scheduling and policing on routers or switches in the path. In admission control, this mechanism determines whether a new flow can be accepted and rejects the application request depending on existing reservations. In classification, the IntServ architecture classifies classes of service such as guaranteed service class, best-effort service class, and controlled-load service class. The application's requesting QoS services such as reserving bandwidth, bounded delay and no-loss assurance are controlled by guaranteed service class. In queuing and scheduling, the packet scheduler mechanism manages the forwarding of packets according to those QoS requests using a set of queues such as timers. In policing, the packet classifier mechanism is used to control traffic, including likely dropping packets, when traffic does not adapt to its required QoS components (Braden et al., 1994).

The main advantage of the IntServ architecture is that the network is able to provide end-to-end QoS with guarantee over throughput and delay. However, it has a scalability problem because it is based on individual flows. It also needs elementary modification in the network core, because all routers along the traffic path have to support it.

2.2.2 Differentiated Services

DiffServ is a computer networking architecture that provides the technique to deliver end-to-end QoS for individual flows. To establish and maintain a QoS, the flows of a packet's class can be marked in the packet directly. In this architecture, packets are classified and marked according to predefined rules to receive specific forwarding action appropriate per-hop behaviour (PHB) on routers in their path. DiffServ produces good QoS scalability in terms of implementing functions such as classifying, marking, traffic shaping and policing at a DiffServ-capable host (Blake et al., 1998). A DiffServ-capable router has to implement packet classification and packet marking on the edge router with the arriving packets marked and classified and all routers have to implement PHBs such as priority queuing and traffic shaping. PHB can only offer soft guarantees along with statistical end-to-end delay. The main uses PHBs are expedited forwarding (EF), assured forwarding (AF), class selector (CS) and default PHB. EF is dedicated lowloss, low-latency traffic (e.g. video). It is implemented using priority queuing along with rate limiting on a traffic class (Jacobson, Nichols, & Poduri, 1999). AF provides assurance of delivery under prescribed conditions (Heinanen, Baker, Weiss, & Wroclawski, 1999). CS maintains backward compatibility with the IP precedence field for network nodes implementing IP precedence-based classification and forwarding and default PHB is typically for best-effort traffic (Nichols, Blake, Baker, & Black, 1998).

The main advantage of DiffServ is that it does not require advance setup, reservation and time consuming end-to-end negotiation for each flow and it allows flexible class definitions and a differentiated pricing of Internet service. Because DiffServ-aware routers apply PHB to traffic classes, owing to lack of reservation it is difficult to predict the end-to-end behaviour of the network. Moreover, DiffServ does not provide hard guarantees for QoS requirements (e.g. delay and throughput).

2.2.3 Multi Protocol Label Switching

MPLS is a label based switching mechanism where the packets are delivered in between source and destination with labels instead of using hop-by-hop IP based delivering. The MPLS network has the facility of switching packets with specific features such as carrying data of particular application types in an assured mode. In MPLS, the short labels are assigned to packets that exclude intricate routing table lookups. The key mechanism is that the labels offer a way of attaching supplementary information to each packet (Filsfils et al., 2009).

In MPLS, a packet is labelled by assigning the packet to a forwarding equivalent class (FEC). This is done only once when an unlabelled packet enters to an MPLS domain. The routing is performed by label-switched routers (LSRs) and label packets are routed according to label-switched paths (LSPs). Route (LSP) selections can be determined by hop-by-hop routing or explicit routing. Hop-by-hop routing allows each LSR to independently choose the next hop for each FEC. In addition, explicit routes can be determined by specific LSRs via source routing or by the network operator manually. Explicit routing is useful for policy routing or traffic engineering (Rosen & Rekhter, 1999).

When an unlabelled packet comes into the MPLS network an MPLS label is attached using FEC. This is done only once. The routing is established in a predetermined path such as LSPs to labelled packets in an MPLS network, based on the rule in FEC. Route (LSP) selections can be determined by hop-by-hop routing or explicit routing. Hop-by-hop routing allows each LSR to independently choose the next hop for each FEC. In addition, explicit routes can be decided by particular LSRs using the source routing technique or by the network administrator manually. Explicit routing is helpful for policy routing or traffic engineering (Rosen, Viswanathan, & Callon, 2001).

MPLS networks have the ability to reduce distortions of streams by setting up multiple LSPs or tunnels between the source and destination to ensure the logical separation between streams (Srivastava, van de Liefvoort, & Medhi, 2009).

The main advantage is complex routing table lookups and MPLS provides ultrafast switching capabilities and is suitable for delay-sensitive applications. Moreover, MPLS is interoperated with both packet-switched and circuit-switched networks. However, it lacks real-time reconfigurability and adaptivity.

The QoS related research gaps in current Internet architecture is shown in the Table 2.1.

Table 2.1: QoS related research gaps in Current Internet Architecture

	IntServ	DiffServ	MPLS	SDN
Software Design	X	X	X	✓
Load Balancing without Server	X	X	X	✓
Per-Flow-Based Network Routing	X	X	X	✓
Centralised Control	X	X	X	✓

2.3 QoS Requirements of Video Streaming Applications

Owing to its real-time nature, video streaming has timing constraints. For example, video and audio data must be played out continuously. If the data do not arrive in time the playout will pause, which is annoying to human ears and eyes. As such, video streaming has specific requirements that need to be met in terms of bandwidth, delay and packet loss. In video streaming, raw video and audio data are compressed by video compression and audio compression algorithms and saved in storage devices. The streaming server retrieves compressed audio and video data from storage devices and sends them to the client. During the process to transmit the data, the transport protocol packetises the compressed bit streams and sends the audio video packets through the Internet. For packet delivered to the receiver they pass through the transport layers and are then processed by the application layer before being decoded at the audio video decoder. The video streaming system is depicted in Figure 2.2.

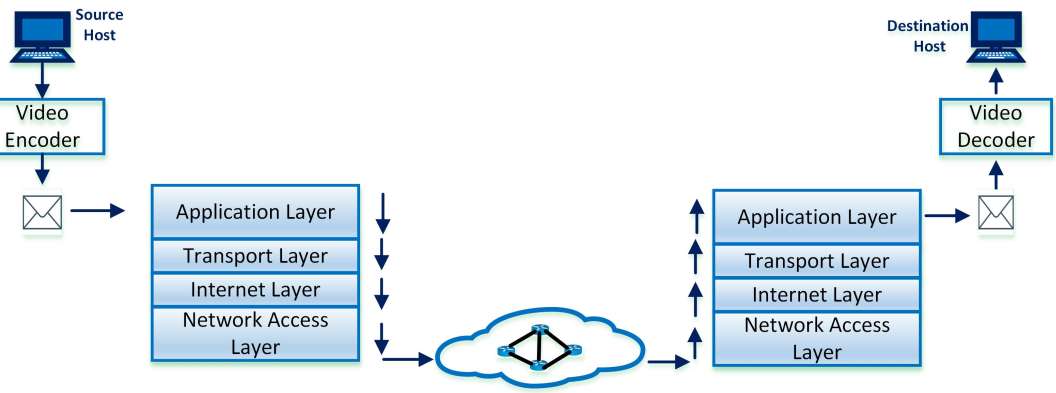

Figure 2.2: Video Streaming System

Video streaming applications represent the video which is the absence of conversational elements. Users can send requests to view the live streaming. The characteristics of video streaming applications are elastic and variable rate. These applications have more tolerant QoS requirements because the delay requirements are not so stringent. However, streaming videos may contain important information such as multicast company meetings which require service guarantees. The QoS requirements of interactive video and streaming video are described in Table 2.2. (W. C. Lee et al., 1995) (Y. Chen, Farley, & Ye, 2004). The value indicated in the table recommend the following:

• The streaming video bandwidth requirement is related to the encoding format and the rate of the video stream. Hence, it is not a fixed value.

• A basic video streaming delay is influenced by the buffering capabilities of the video application. Thus, it can be less than the value indicated.

• The tolerable jitter is not an essential parameter in video streaming applications. Thus, there are no significant jitter requirements.

Table 2.2: QoS Requirements of Streaming Video and Interactive Video (Y. Chen et al., 2004)

Medium	Application	Degree of Symmetry	Delay	Packet Loss	Jitter	Band width
Streaming Video	Surveillance, Movie Clips, Real Time Video	Primarily One-Way	<= 5s	<= 5	Not Applicable	Not Applicable
Interactive Video	Video Conferencing	Primarily One-Way	<=150 ms	<=1	<=30 ms	Not Applicable

In multimedia applications, the QoS also depends on the video codec type used, the error recovery mechanism and the content bit rates.

There are several video coding techniques that are widely used for video delivery, namely scalable video coding (SVC) and MDC. These techniques are used because of their robustness during transmission especially video streaming applications.

2.3.1 Scalable Video Coding

SVC is a coding method for modern video transmission systems. In this approach, the video bit stream consists of one or more subset bit streams that can adapt to the different requirements or options of end users, including changing network conditions and could decode themselves to reconstruct quality video. In other words in SVC, the video is encoded as one base layer which can be decoded individually and as several enhancement layers, which can be decoded cumulatively. These layers distribute different qualities with various bandwidth characteristics or when the network characteristics are time varying. The video server encodes video by three dimensions to several layers, and the layers consist of one base layer and one or several enhancement layers, as described in Figure 2.3. The base layer has a single video stream with a certain frame rate, and the temporal scalability adds enhancement layers with additional frame rates. The base layer provides basic video quality and the enhancement layer is used to refine the base layer video quality as shown in Figure 2.4.

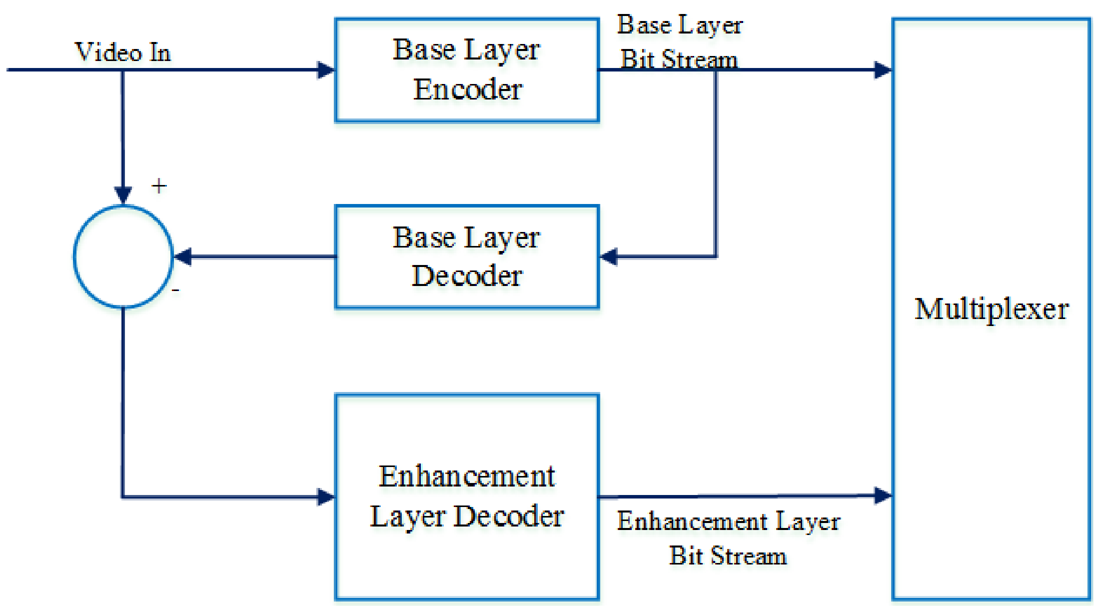

Figure 2.3: Encoding Video into Several Layers(Y.-C. Lee et al., 2003)

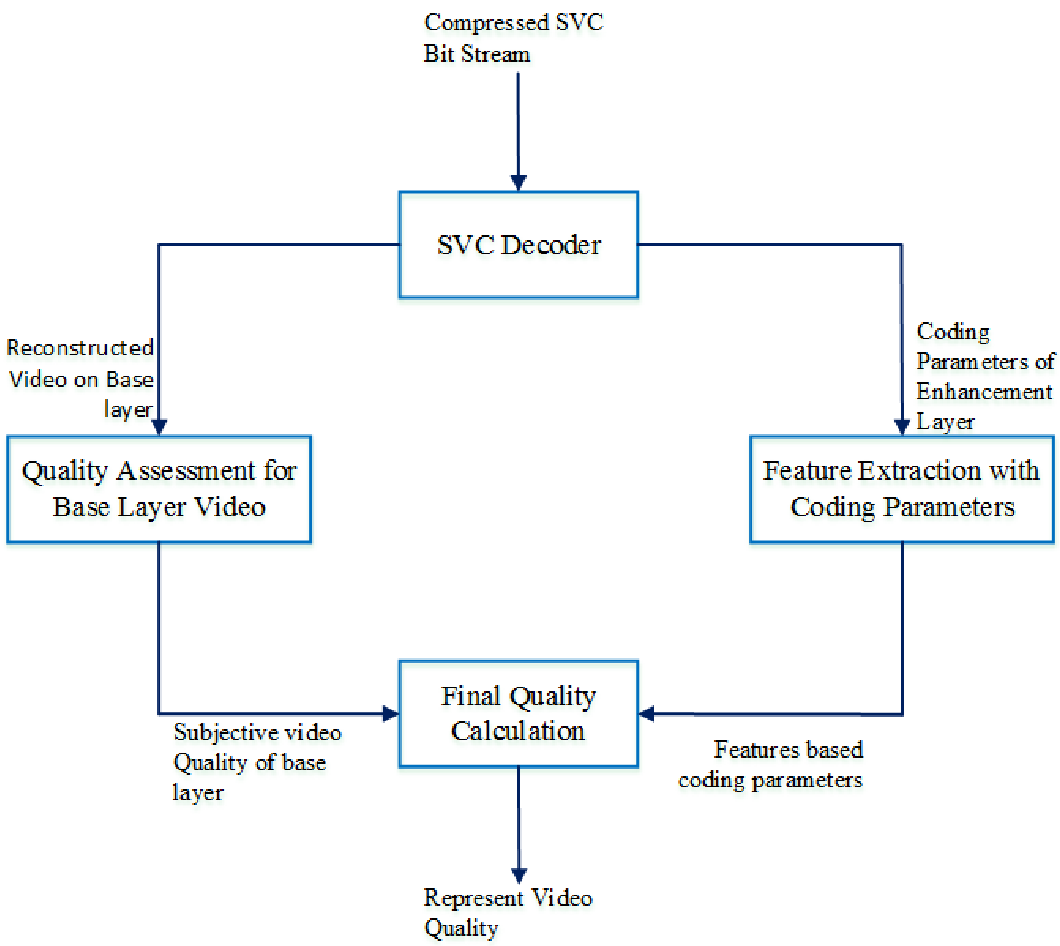

Figure 2.4: Decoding Video with Additional Frame Rates(Y.-C. Lee et al., 2003)

(Fallah, Mansour, Khan, Nasiopoulos, & Alnuweiri, 2008) and (H. Mansour, 2008) suggested a technique that produces two descriptions for each enhancement layer of an SVC coded stream by embedding in each description only half of the motion and texture information of the original coded stream with a minimal degree of redundancy. The generated descriptions are interdependent and independently decodeable. Moreover, if only one description is received, then the missing motion data can be recovered using a decoder and produce an output video sequence with good enough quality. On the other hand, if two descriptions are received, then the full quality of a single description SVC stream is delivered. The suggested framework offers a highly error-resilient video bit stream that needs no retransmissions or feedback channels while minimising any channel overhead imposed by the video redundancy due to multiple description coding.

The main advantage of SVC is that part of the scalable video stream is sent or decoded at the destination side, depending on the available network resources and displaying capabilities. On the other hand, when the base layer data are lost, the decoder cannot decode even though the enhancement data are received.

2.3.2 Multiple Description Coding (MDC)

The MDC approach is a video coding method that enhances the resilient coding and transmission for image and video signals. An MD coder encodes a media source into two or more bit streams where each description should contain a sufficient amount of information about the original source. That is, a certain amount of correlation has to be adopted to help reconstruct signals from each description. These bit streams, also called descriptions, are generated in such a way that each description can be independently decoded to produce a signal of basic quality. Thus, MDC technique provides a quality image or video in the presence of packet losses during transmission. In general, a sequence can be divided into multiple descriptions by MDC based on three domains: spatial, temporal and frequency (Heng, Apostolopoulos, & Lim, 2006)

An MD video codec for two descriptions is shown in Figure 2.5. If description D1 is received, the side decoder reconstructs the source with a higher but acceptable distortion. When both descriptions are received the central decoder reconstructs the original source D0 clearly, from D0 D1 and D0 D2.

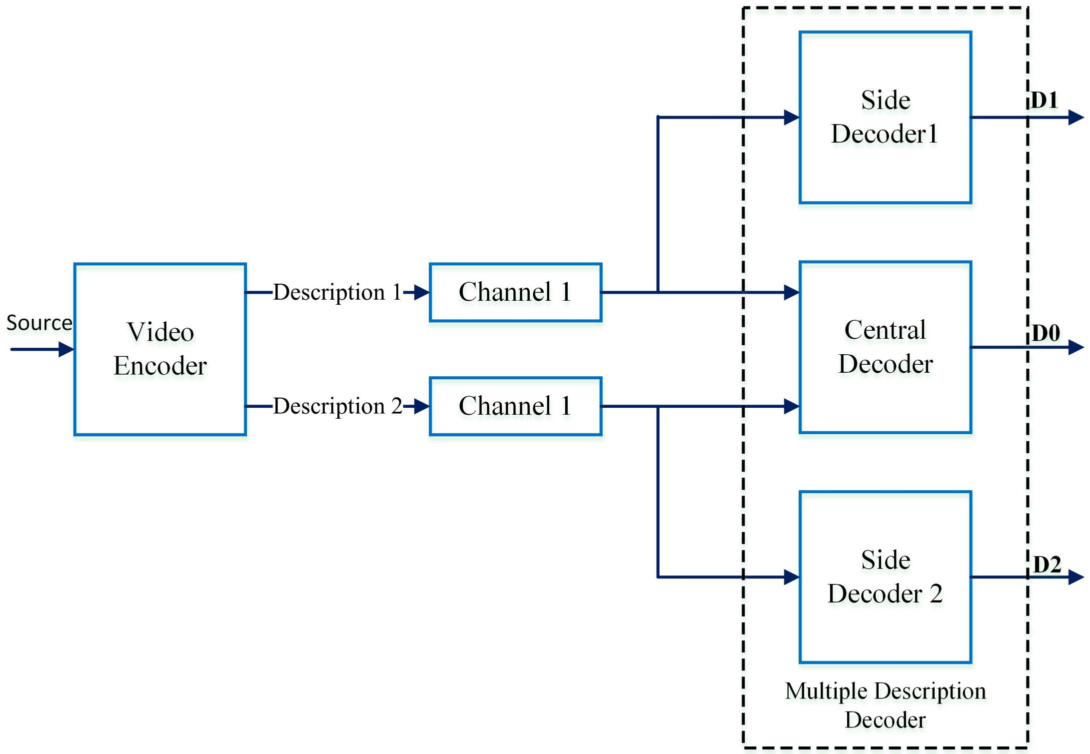

Figure 2.5: MD Video Codec for Two Descriptions (Y.-C. Lee et al., 2003)

(Bajic & Woods, 2002) proposed a wavelet transform coefficients-based MDC scheme called domain-based MDC, where the redundancy of information is avoided. As the missing coefficient is estimated from its immediate neighbours the spatial redundancy in the transform domain of the signal is used to construct lost descriptions from the received ones. This only focuses on finding the best way to partition the transformed coefficients but ignores adopting the inherent correlation within the signals to reconstruct the corrupted coefficients at the decoder. The main advantage of MDC is that when streaming the video, any lost description does not affect the decoding of other descriptions. In the packet loss scenario the loss increases (by more than 5%) and MDC performs better than SVC. In a real- time or live video applications, the decoding deadline is not acceptable when the retransmission of a lost packet is missed. Under such conditions, the MDC approach is the appropriate choice for streaming. Example: The strict demanding requirements of high definition video conferencing

(HDVC), such as maximum delay threshold and minimum quality threshold make it a challenging application for video streaming.

2.4 Software-Defined Networking

SDN is an emerging networking paradigm that separates the network control plane from the data forwarding plane with the promise to dramatically improving network resource utilization, simplifying network management, reducing operating costs and promoting innovation and evolution. The SDN architecture can be represented by three different logical layers as shown in Figure 2.6 (SDN, 2012).

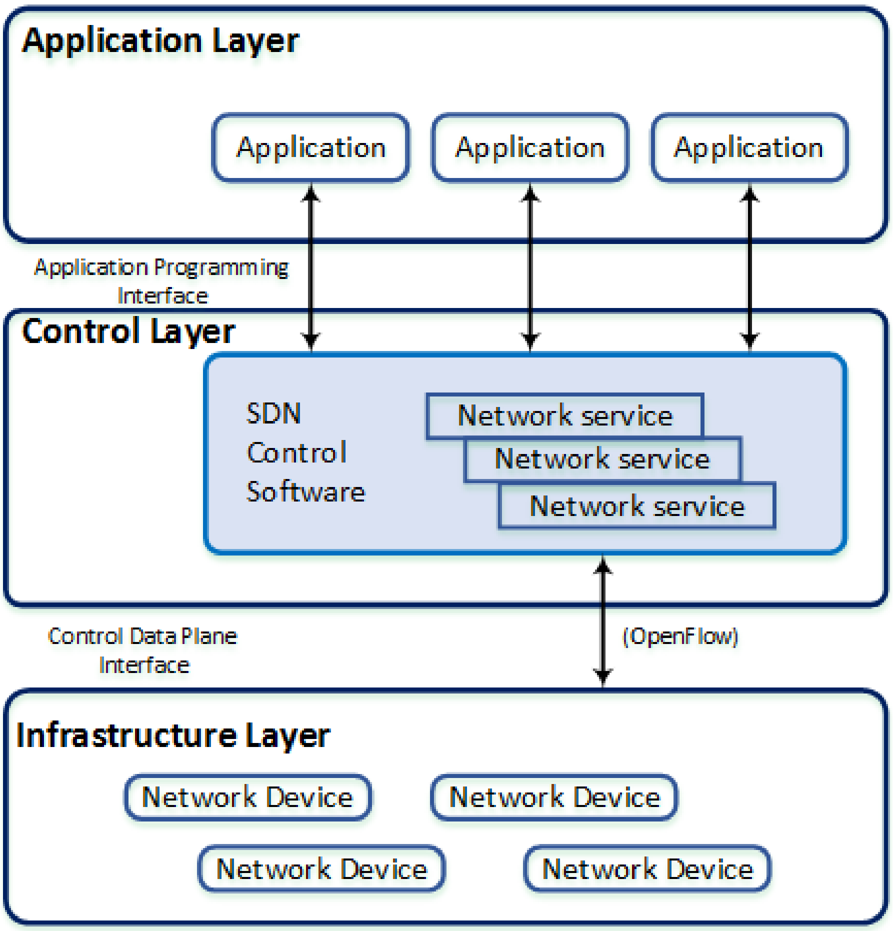

Figure 2.6: SDN Architecture